IN THE CLAIMS:

This listing of claims will replace all prior versions, and listings, of claims in the application:

LISTING OF CLAIMS:

1. (original) A method, comprising the steps of:

(a) obtaining information relevant to the quality of service of voice calls being transmitted from a first location to a second location via an IP network;

(b) calculating a parameter based on said information; and

(c) accepting a new call into the IP network in the case of said parameter not exceeding an upper threshold.

2. (original) The method of claim 1 wherein said new call is accepted into the IP network at a reduced bandwidth in the case of said parameter exceeding a lower threshold.

3. (original) The method of claim 1 where said new call is not accepted into the IP network in the case of said parameter exceeding the upper threshold.

4. (previously presented) The method of claim 1 wherein the information obtained is a number of sent packets transmitted from said first location to said second location in the IP network, wherein the number of sent packets comprises a number of lost packets, a number of late packets and a number of received packets.

5. (original) The method of claim 1 wherein the information obtained is a delay of received packets transmitted from said first location to said second location in the IP network.

6. (original) The method of claim 1 wherein the information obtained is a delay variation of received packets transmitted from said first location to said second location in the IP network.

7. (original) The method of claim 1 wherein the information is obtained on a periodic basis.

8. (original) The method of claim 1 wherein the information is obtained on an exception basis using an immediate report.

9. (original) The method of claim 1 wherein the parameter is identified as a packet lost ratio (PLR).

10. (original) The method of claim 9 wherein PLR is defined as

$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}.$$

11. (original) The method of claim 2 wherein bandwidth is reduced for a newly accepted call by selecting a first encoder to encode the new voice call information in a bandwidth that is smaller than bandwidths of other calls accepted in the network that are encoded by a second encoder.

12. (previously presented) The method of claim 2 wherein the bandwidth of a newly accepted call is reduced by increasing the packet size for said newly accepted voice call, wherein the packet size is indicative of a size of a corresponding voice sample.

13. (original) The method of claim 2 wherein the bandwidth of a newly accepted call is reduced by activating the characteristic of silence suppression for said newly accepted voice call.

14. (original) Apparatus comprising a gateway for interfacing voice call data from a public switch telephone network to an internet protocol network; said gateway further comprising:

a first circuit for passing said voice call data to the internet protocol network;

a second circuit for polling the internet protocol network about traffic information transmitted therein; and

a third circuit for processing the polled information to determine whether the voice call data is to be accepted by the internet protocol network.

15. (original) The apparatus of claim 14 wherein said first circuit further comprises one or more Ethernet cards that are connected to the internet protocol network.

16. (original) The apparatus of claim 14 wherein said second circuit is at least one strongarm card.

17. (original) The apparatus of claim 16 wherein the strongarm card is connected to the Ethernet card via a host CPU circuit.

18. (original) The apparatus of claim 14 wherein the third circuit compares a parameter based on the polled information to a plurality of thresholds.

19. (original) The apparatus of claim 18 wherein the parameter is a packet loss ratio defined as

$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}.$$

20. (original) The apparatus of claim 19 wherein the third circuit compares the packet loss ratio to a lower threshold and if the packet loss ratio is less than the lower threshold, a new voice call is accepted into the internet protocol network.

21. (original) The apparatus of claim 19 wherein the third circuit compares the packet loss ratio to the lower threshold and an upper threshold, and if lower threshold < packet loss ratio < upper threshold, a new voice call is accepted into the internet protocol network at a reduced bandwidth.

22. (original) The apparatus of claim 19 wherein the third circuit compares the packet loss ratio to the upper threshold, and if the packet loss ratio is greater than the upper threshold, a new voice call is blocked from entering the internet protocol network.